DETAILED ACTION

Response to Amendment

1. Claims 2-7 and 9 are pending.

2. Claim 2 has been amended.

3. Claims 1 and 8 have been canceled.

Continued Examination Under 37 CFR 1.114

4. A request for continued examination under 37 CFR 1.114, including the fee set forth in 37 CFR 1.17(e), was filed in this application after final rejection. Since this application is eligible for continued examination under 37 CFR 1.114, and the fee set forth in 37 CFR 1.17(e) has been timely paid, the finality of the previous Office action has been withdrawn pursuant to 37 CFR 1.114. Applicant's submission filed on 9/1/2009 has been entered.

Response to Arguments

5. Applicant argues "Applicant respectfully submits that DeJaco does not teach or suggest a method of preprocessing audio data by using the same rate decision algorithm as that of the predetermined codec before subjecting the audio data to the predetermined codec such that, in the overall system (including preprocessing and encoding), the rate decision algorithm is used twice." (Remarks, Page 9, ¶ 1) In response to applicant's argument that the references fail to show certain features of applicant's invention, it is noted that the features upon which applicant relies (i.e.,

...providing the preprocessed frames of audio data to the predetermined codec,...is decided by the predetermine rate decision algorithm...) are not recited in the rejected claim(s). Although the claims are interpreted in light of the specification, limitations from the specification are not read into the claims. See *In re Van Geuns*, 988 F.2d 1181, 26 USPQ2d 1057 (Fed. Cir. 1993).

6. Applicant further argues "Malvar appears to simply describe automatic gain control without any detailed function, and does not teach or suggest analyzing audio data by using the rate decision algorithm of the predetermined codec, so as to select the frames that are classified as noise data if provided to the predetermined codec." (Remarks, Page 9, ¶ 4) In response to applicant's argument that the references fail to show certain features of applicant's invention, it is noted that the features upon which applicant relies (i.e., ...providing the preprocessed frames of audio data to the predetermined codec,...is decided by the predetermine rate decision algorithm...) are not recited in the rejected claim(s). Although the claims are interpreted in light of the specification, limitations from the specification are not read into the claims. See *In re Van Geuns*, 988 F.2d 1181, 26 USPQ2d 1057 (Fed. Cir. 1993).

7. Applicant further argues "Applicant respectfully submits that even assuming arguendo, the combination of DeJaco and Malvar merely creates a modified predetermined codec with automatic gain control, which is distinguishable from the claimed invention which does not require any modification to the predetermined codec itself or rate decision algorithm in the predetermined codec, as previously discussed." (Remarks, Page 9, ¶ 5) This argument is moot in view of the new ground of rejection

provided by Benyassine et al. Furthermore, the Examiner notes that the limitation addressed in the argument (does not require modification) was not present in the previously filed claim language, therefore although the claims are interpreted in light of the specification, limitations from the specification are not read into the claims. See *In re Van Geuns*, 988 F.2d 1181, 26 USPQ2d 1057 (Fed. Cir. 1993).

8. Applicant further argues "Applicant respectfully submits that the combination of the references for the purposes of the present rejection is improper because of the failure of either patent to suggest the combination. It is a requirement that in making a combination of patents in a rejection, those patents must suggest the desirability of the combination of teachings. This requirement was expressed by the Court of Customs and Patent Appeals in In re Imperato, 179 U.S.P.Q. 730..." (Remarks, Page 12, ¶ 2) The Examiner disagrees. Malvar, column 2, lines 41-56 teaches the use of automatic gain control as a pre-processing step. One of ordinary skill in the art could have looked to Malvar to provide an automatic gain control step as a pre-processing step to DeJaco. However, Hardiman et al. is provided as a replacement to Malvar because it more expressly teaches the automatic gain control methods.

Specification

9. The disclosure is objected to because of the following informalities: ¶ 0025 - ¶ 0027 and ¶ 0029 of the specification show a character (small box shape) that is not present in equation 3, to which it is referenced. Appropriate correction is required. The Examiner respectfully requests that applicant check the remainder of the specification

for similar errors.

10. The disclosure is objected to because of the following informalities: Equation 3 in ¶ 0025 should have a zero (0) instead of an "o" in the calculation for beta. Appropriate correction is required.

11. The disclosure is objected to because of the following informalities: Table 4 (page 19) incorrectly spells Beethoven's sonata as "Sonata Pathetic." It should be spelled "Sonata Pathetique" Appropriate correction is required.

12. The disclosure is objected to because of the following informalities: ¶ 0069 misspells VoiceXML as "VoiceXLM." Appropriate correction is required.

13. Claim 1 is objected to because the following limitations are not given patentable weight. The limitation "so as to select at least some frames that, when ... by the predetermined codec." is not given patentable weight. Quoting Minton v. Nat'l Ass'n of Securities Dealers, Inc., 336 F.3d 1373, 1381, 67 USPQ2d 1614, 1620 (Fed. Cir. 2003)) that a "'whereby clause in a method claim is not given weight when it simply expresses the intended result of a process step positively recited.'" Id

Claim Rejections - 35 USC § 103

The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

14. Claims 2, 4, and 9 are rejected under 35 U.S.C. 103(a) as being unpatentable over DeJaco (US Pat 5742734) in view of Hardiman (US Pat 5937377) and further in view of Benyassine (US Pat 6694293).

As per claim 2, DeJaco teaches the method comprising:

for each frame, deciding, in the computing system, an encoding rate for that frame using the predetermined rate decision algorithm of the predetermined codec, so as to select at least some frames that, when provided to the predetermined codec, are classified as noise data and encoded at the lowest encoding rate by the predetermined codec; (DeJaco, column 1, lines 12-22, …*The rate determination algorithm assigns a higher bit rate encoding scheme to segments of the audio signal in which speech is present and a lower rate encoding scheme for silent segments…* DeJaco more specifically teaches the rate selection on column 3 lines 56-67 to column 4, lines 1-15 based on the energy in each subband. Column 5, lines 13-64 further teaches that the thresholds are developed based on the estimated signal to noise ratio, which detects voice (column 2, lines 24-33).)

DeJaco fails to fully teach, but Hardiman teaches:

adjusting, in the computing system, energy of the at least some frames of audio data selected in the deciding so as to produce preprocessed frames of audio data that, when provided to the predetermined codec, are classified as valid voice data by the predetermined rate decision algorithm and encoded at the one of the plurality of

encoding rates other than the lowest encoding rate thereof; and (DeJaco provides classification of voice data through voice detection thresholds (column 2, lines 24-33). DeJaco does not explicitly state that the voice data is not encoded at the lowest rate, but as shown in Table 1 (column 5, lines 31-40) SNR values limit the rate thresholds. Therefore, a high SNR indicating a voice detection would not be encoded at the lowest rate. Hardiman, abstract and claim 1 teaches automatic gain control as a pre-processing step for the reasons set forth in column 12, lines 16-24.)

wherein preprocessing the at least some frames of audio data causes the predetermined codec to classify the preprocessed frames of audio data as valid voice data instead of noise data. (DeJaco teaches that previous speech coding systems do not correctly determine when low energy unvoiced speech is input (column 1 lines 40-52). The systems often mistake low energy unvoiced speech as noise and encode the signal at a lower bit rate, causing degradation in speech quality during speech reconstruction (column 1 lines 40-52). Hardiman, abstract and claim 1 teaches automatic gain control as a pre-processing step for the reasons set forth in column 12, lines 16-24 and further states that background noise is reduced (abstract). It would have been obvious considering the application of Hardiman to use automatic gain control on the audio data of DeJaco to more easily distinguish voice and noise.)

It would have been obvious to someone of ordinary skill in the art at the time of the invention to combine Hardiman with DeJaco to “compensate for the overall gain variation of the input signal to a desired level” (column 12, lines 16-24) Additionally, the combination of Hardiman with DeJaco provides automatic gain control and adaptive

equalization for IS-127 (EVRC). (column 6, lines 34-43)

DeJaco and Hardiman fail to fully teach, but their combination with Benyassine teaches:

providing the preprocessed frames of audio data to the predetermined codec, so that in the predetermined codec, the encoding rate of the preprocessed frame is decided by the predetermined rate decision algorithm; (The combination of DeJaco and Hardiman teaches two classification processes, one before and one after the gain control. This can be seen in Hardiman, column 6, lines 3-17, where a known classification of speech or noise is known for controlling the gain. DeJaco determines the rate by the signal to noise ratio, which detects voice in column 2, lines 24-33. However, DeJaco and Hardiman fail to teach that the encoding rate of the preprocessed frame is decided by the predetermined rate decision algorithm. Since Hardiman provides a classification prior to the gain control, it would have been obvious that the rate decision algorithm could have been run because the rate decision algorithm determines the rate based on the classification of the signal. The classification of the signal is ultimately what determines the rate, and the classification is used for gain control in Hardiman. Benyassine, Fig .2, teaches a classification for noise, music, and speech which adjusts the bit rate of the signal depending on the signal classification (column 2, lines 22-43). DeJaco lastly provides the final rate selection based on a final classification. The two-stage classification of Benyassine could have been applied to DeJaco and Hardiman to provide classification before automatic gain control and afterwards.

It would have been obvious to someone of ordinary skill in the art at the time of the invention to combine Benyassine with DeJaco and Hardiman because "the speech coding of the music frame may be done at higher bit rates to accommodate the music signal" (column 9, lines 43-51) The combination of Benyassine with DeJaco and Hardiman provides that the speech coding of a music frame could be performed at higher bit rates after adjustment by automatic gain control to better establish that the signal comprises music and not speech or noise.

As per claim 4, claim 2 is incorporated and DeJaco fails to teach but Hardiman teaches

calculating signal levels of the selected frames of the audio data;

(Column 9, lines 1-9, *...In its estimation of the current input energy, the AGC 203 utilizes the noise reducer parameters P.sub.NR determined during the conventional noise reduction process. These parameters include the frame counter (frame.sub.-- number), the noise frame indicator (update.sub.-- flag), the total channel energy E.sub.TOT (m), and the current channel energy E(m,i). It should be noted that the modified gain computation by the AGC 203 is implemented only when speech signal is present...*)

determining gain values based on the calculated signal levels produced by the calculating; and (Column 9, lines 1-9, *...In its estimation of the current input energy, the AGC 203 utilizes the noise reducer parameters P.sub.NR determined during the conventional noise reduction process. These parameters include the frame counter (frame.sub.-- number), the noise frame indicator (update.sub.-- flag), the total channel*

energy E.sub.TOT (m), and the current channel energy E(m,i). It should be noted that the modified gain computation by the AGC 203 is implemented only when speech signal is present…)

generating preprocessed frames of audio data by multiplying the gain values to the selected frames of audio data. (column 12, equation 5, fig. 2 (205))

It would have been obvious to someone of ordinary skill in the art at the time of the invention to combine Hardiman with DeJaco to "compensate for the overall gain variation of the input signal to a desired level" (column 12, lines 16-24) Additionally, the combination of Hardiman with DeJaco provides automatic gain control and adaptive equalization for IS-127 (EVRC). (column 6, lines 34-43)

As per claim 9, claim 2 is incorporated and DeJaco fails to explicitly teach, but Hardiman teaches:

wherein the computing system for preprocessing audio data is a separate system from the predetermined codec. (Hardimas perform classification and AGC prior to coding as preprocessing steps. Therefore it would have been obvious to someone of ordinary skill in the art at the time of the invention that the preprocessing steps could have been performed independently of the variable rate coder such as, for example, a distributed coder.)

It would have been obvious to someone of ordinary skill in the art at the time of the invention to combine Hardiman with DeJaco to "compensate for the overall gain variation of the input signal to a desired level" (column 12, lines 16-24) Additionally, the

combination of Hardiman with DeJaco provides automatic gain control and adaptive equalization for IS-127 (EVRC). (column 6, lines 34-43)

15. Claim 3 is rejected under 35 U.S.C. 103(a) as being unpatentable over DeJaco (US Pat 5742734) in view of Hardiman (US Pat 5937377) and further in view of Benyassine (US Pat 6694293) and further in view of Eryilmaz (US Pat 5867574)

As per claim 3, claim 2 is incorporated and DeJaco and Hardiman fail to fully teach, but Eryilmaz teaches:

further comprising the step of determining whether a frame in the audio data is a silence frame based on the energy of the frame, wherein when the frame is a silence frame, the energy thereof is not adjusted in the adjusting step.

(DeJaco, column 2 lines 15-18 and lines 39-42, the input signal is analyzed to determine the presence of speech or music. If the input signal is neither speech nor music, then it must be noise or silence with background noise. Further, Eryilmaz, claim 22, *…said automatic gain control means communicating with said voice activity detector to insert gain only upon the detection of said voice activity…*)

It would have been obvious to someone of ordinary skill in the art at the time of the invention to only provide gain to known voiced frames to avoid boosting background noise into the signal. (column 9, lines 41-47)

16. Claims 5-7 are rejected under 35 U.S.C. 103(a) as being unpatentable over

DeJaco (5,742,734) in view of Hardiman (US Pat 5937377) and further in view of Benyassine (US Pat 6694293) and further in view of Claesson et al. (US Pre-Grant Publication # 20030023429).

As per claim 5, claim 4 is incorporated and DeJaco fails to teach, but Hardiman teaches:

wherein the frame includes a set of samples including a current sample, and a signal level for the current sample is determined based on the current sample and other samples adjacent to the current sample,

(Column 9, lines 1-9, *...In its estimation of the current input energy, the AGC 203 utilizes the noise reducer parameters P.sub.NR determined during the conventional noise reduction process. These parameters include the frame counter (frame.sub.--number), the noise frame indicator (update.sub.-- flag), the total channel energy E.sub.TOT (m), and the current channel energy E(m,i). It should be noted that the modified gain computation by the AGC 203 is implemented only when speech signal is present...* the updated values take adjacent samples into consideration.)

It would have been obvious to someone of ordinary skill in the art at the time of the invention to combine Hardiman with DeJaco to “compensate for the overall gain variation of the input signal to a desired level” (column 12, lines 16-24) Additionally, the combination of Hardiman with DeJaco provides automatic gain control and adaptive equalization for IS-127 (EVRC). (column 6, lines 34-43)

Bhaskar fails to teach, but Claesson teaches:

the gain value for the current sample in the frame is determined based on the signal level of the current sample. (Claesson, ¶ 0048, ... *The future sample is multiplied by the gain factor. If the resulting data has an amplitude greater than a threshold value (a user-fixed parameter) the gain factor is reduced to a value equal to the threshold value divided by the amplitude of the future sample...Finally, the sample in the buffer which has been delayed is multiplied by the gain factor described above in order to produce the output...*)

It would have been obvious to someone of ordinary skill in the art at the time of the invention to combine Claesson with DeJaco, Hardiman, and Benyassine because (¶ 0007) undesirable artifacts are generated from low bit rate encoding schemes and Claesson provides automatic gain control which depend on certain parameters that can be selected depending on the application and desired effect. (¶ 0012) Therefore it would have been obvious to combine Claesson with DeJaco, Hardiman, and Benyassine to use automatic gain control tailored to the application.

As per claim 6, claim 5 is incorporated and DeJaco, Hardiman, and Benyassine fail to teach, but Claesson teaches:

wherein the signal level for the current sample is determined based on the current sample and a first set of samples within an attack time ahead of the current sample, and a second set of samples within a release time behind the current sample. (Claesson, ¶ 0072, ...*the AGC blocks of the present invention examine the recent*

history and/or immediate future of the signal and use this information to adjust a gain factor such that the signal is kept within a range of peak excursion…)

It would have been obvious to someone of ordinary skill in the art at the time of the invention to combine Claesson with DeJaco, Hardiman, and Benyassine because (¶ 0007) undesirable artifacts are generated from low bit rate encoding schemes and Claesson provides automatic gain control which depend on certain parameters that can be selected depending on the application and desired effect. (¶ 0012) Therefore it would have been obvious to combine Claesson with DeJaco, Hardiman, and Benyassine to use automatic gain control tailored to the application.

As per claim 7, claim 6 is incorporated and DeJaco, Hardiman, and Benyassine fail to teach, but Claesson teaches:

wherein the attack time and the release time can be changed based on the characteristic of the audio data. (Claesson, ¶ 0012, *...the present invention provides methods and apparatus for effecting automatic gain control for a sampled signal. Specific embodiments are described as algorithms that depends on certain parameters that can be selected depending on the application and the desired effect…*)

It would have been obvious to someone of ordinary skill in the art at the time of the invention to combine Claesson with DeJaco, Hardiman, and Benyassine because (¶ 0007) undesirable artifacts are generated from low bit rate encoding schemes and Claesson provides automatic gain control which depend on certain parameters that can be selected depending on the application and desired effect. (¶ 0012) Therefore it would

have been obvious to combine Claesson with DeJaco, Hardiman, and Benyassine to use automatic gain control tailored to the application.

Conclusion

17. The prior art made of record and not relied upon is considered pertinent to applicant's disclosure. Refer to PTO-892, Notice of References Cited for a listing of analogous art.

18. Any inquiry concerning this communication or earlier communications from the examiner should be directed to GREG A. BORSETTI whose telephone number is (571)270-3885. The examiner can normally be reached on Monday - Thursday (8am - 5pm Eastern Time).

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, RICHEMOND DORVIL can be reached on 571-272-7602. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

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/Greg A. Borsetti/
Examiner, Art Unit 2626

/Talivaldis Ivars Smits/
Primary Examiner, Art Unit 2626 10/30/2009